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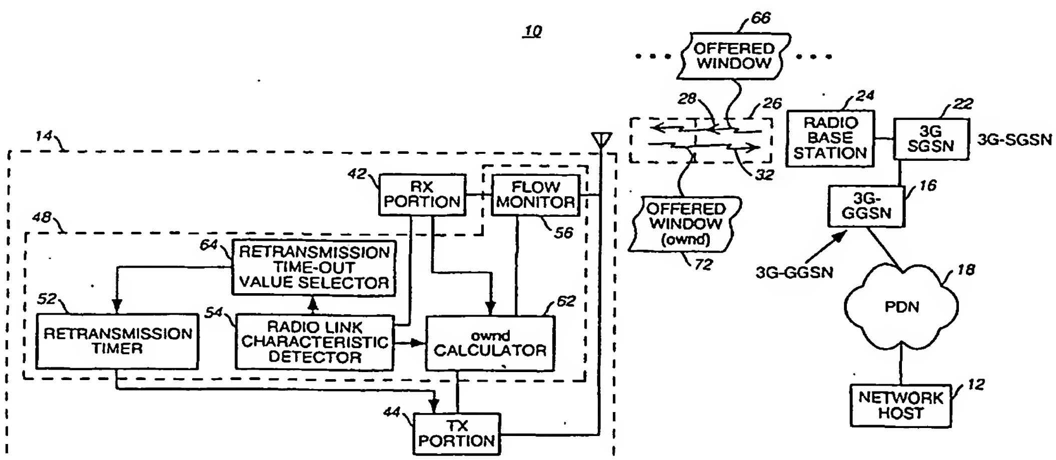
(54) Title: APPARATUS, AND ASSOCIATED METHOD, FOR COMMUNICATING PACKET DATA IN A NETWORK INCLUDING A RADIO-LINK



(57) Abstract: Apparatus (48), and associated method, for improving packet data communications upon a communication path including a radio-link (26). Determination is made of the conditions on the radio-link (26) when selecting the optimal size of a transmission window (62) within which to transmit packets of data. And, retransmission time-out values (64) are also selected responsive to the indications of the radio-link conditions (54).

WO 01/93513 A2

APPARATUS, AND ASSOCIATED METHOD, FOR COMMUNICATING PACKET DATA IN A NETWORK INCLUDING A RADIO-LINK

The present invention relates generally to the communication of packet data, such as TCP-formatted data, in a communication system which includes a radio-link, such as a UMTS (universal mobile telephone service) wireless data network. More particularly, the present invention relates to apparatus, and an associated method, by which more optimally to communicate data packets in the UMTS, or other, communication system.

BACKGROUND OF THE INVENTION

Advancements in communication technologies have permitted the introduction of, and popularization of, new types of communication systems. As a result of such advancements, significant increases in the rates of data transmission, have been permitted. And, new types of communication services have also been made possible.

A radio communication system is exemplary of a type of communication system which has benefited from advancements in communication technologies. At least a portion of a communication path utilized in a radio communication system includes a radio-link. A radio communication system inherently increases communication mobility as communication channels defined in such a system are formed of radio channels and do not require wireline connections for their formation.

Advancements in digital communication techniques are amongst the advancements in communication

technologies which have permitted the introduction of the new types of communication systems. Communications effectuated through the use of digital communication techniques are generally of improved bandwidth efficiencies in comparison to communications effectuated utilizing conventional, analog techniques.

A packet data communication system is also exemplary of a communication system made possible as a result of advancements in communication technologies. In a packet communication system, groups of digital bits are formatted into packets to form packets of data. The packets of data are communicated, either individually, or in groups, at discrete intervals. Once received, the packets of data are concatenated together to recreate the informational content of the digital bits of which the packets are formed.

Because packets of data can be communicated at discrete intervals, the communication channel upon which the packets are transmitted need not be dedicated to a single communication pair. Instead, a shared communication channel can be used by a plurality of communication pairs to communicate packets of data on the shared channel.

Standardized protocols by which to format and to communicate packets of data have been developed. A TCP/IP (transmission control protocol/ Internet protocol) is exemplary of a packet formatting scheme. And, an X.25 protocol describes another exemplary protocol scheme. Standards relating to conventional packet communication systems have been promulgated

for both conventional wireline, as well as wireless, systems.

Packet radio services have been proposed, for instance, for several different cellular communication systems. A cellular communication system is a type of radio communication system, widely implemented and popularly-used. Exemplary of such a packet radio service is the GPRS (General Packet Radio Service) system for GSM (Global System for Mobile Communications).

One proposal is for a so-called 3G (third generation) cellular communication system, referred to as a UMTS (universal mobile telecommunications system) network. Packet data communications are provided for therein. In this proposed system, as well as others, packet data is communicated between a mobile host and a network host. A communication path formed between the mobile and network hosts includes at least one radio-link formed between the mobile host and infrastructure of the UMTS network. Proposals related to the UMTS network include the use of TCP/IP protocols for end-to-end communications, viz., for communications over the wireless and also the fixed parts of the UMTS network. The infrastructure of the UMTS network includes both a wireline IP-based UMTS core network and a radio part, i.e., a radio-link, formed between the mobile host and a base station, forming a portion of the UMTS core network and also over the Internet. TCP-based protocols have, however, conventionally been designed for conventional, wireline networks. Flow and congestion control of a TCP network are designed for

wireline networks where packet losses are often the result of congestion. Congestion arises, for instance, because of the aforementioned sharing of communication resources for different communication pairs. Such service is typically a "best effort" service, i.e., a service without a guaranteed quality of service. When a packet communication system is implemented in wireless form, however, packet losses are often due to bit errors introduced during transmission on a radio-link.

Because a UMTS network includes both a wireline, core network and also a radio part, packet losses occurring at the radio part, such as due to communication handovers or corruption on the radio-link are retransmitted locally. When the UMTS is defined in terms of logical layers, that is, lost packets are retransmitted locally, for example by an RLC (radio-link control) layer. These local retransmissions decrease end-to-end throughput between the mobile and network hosts due to the increased time required to effectuate the local retransmissions. Used in conventional techniques, a sending station which originates TCP data continues sending the packet data at a constant rate, irrespective of the local retransmissions at the radio part of the UMTS network. Thus, congestion of the UMTS core network increases as new packets are transmitted from the network host while earlier packets are still undergoing retransmission to recover from losses in the radio-link. As a result, such additional packet data increases congestion of the UMTS core network. Deleterious results, such as

spurious time-outs of the sending station, might occur which significantly reduces the end-to-end performance of the network.

If a manner could be provided by which better to effectuate when packet data is transmitted by a sending station to take into account for the performance of the radio part of the system, improved system operation would result.

QoS (Quality of Service) levels are also proposed to be defined in the UMTS network. The QoS levels define, in general, performance parameters pursuant to which a particular communication service is to be effectuated. Several communication services are non-real-time services, such as communications with the WWW (World Wide Web), TELNET™, E-MAIL SERVICES, etc. Applications to effectuate such services, logically, run on top of a TCP logical layer. And, such communication services typically are implemented at QoS levels referred to as "best-effort" traffic classes. Such traffic classes do not give guarantees of available bandwidth and, hence, delivery times. Conversely, communication services which are of a real-time nature typically are implemented at higher QoS levels and such communication services are effectuated with a higher priority than non-real-time TCP-related services. Because of the lower priority levels at which the TCP-related, non-real-time services, the bandwidth available to effectuate such services are susceptible to rapid changes.

Conventional manners by which to effectuate TCP flow control do not include a manner by which to set transmission rates according to such rapid changes.

In conventional TCP implementations, a standard mechanism, referred to as self-clocking behavior is used to limit the transmission rate of a sending station. Self-clocking behavior refers to a manner by which the sending station is able to send a new packet, responsive to reception of an acknowledgment of an earlier-transmitted packet, if the size of the transmission window remains constant. In the following description, a segment shall generally refer to a portion of a packet. In conventional TCP operation, however, the transmission window is not constant. Instead, the transmission window is of a size which is adjusted regularly, according to the arrival of acknowledgments and retransmission time-outs. In practice, then, in standard operation, a sending station increases a transmission window size until some point in a communication path, such as an SPSN (Serving Packet Service Node) becomes congested, and data packets are discarded.

In a 3G network, such as a UMTS network, an equivalent network element is known as a 3G-SGSN (third generation, serving GPRS Support Node. Thereafter, congestion control mechanisms are implemented, but such implementations abruptly slow down transmission rates of communications. This behavior also results in reduced end-to-end throughput rates.

If a manner could be provided by which better to effectuate communication of packet data by taking into better account rapid changes of bandwidth availabilities for the communication of the packet data, improved system operation would further result.

It is in light of this background information related to packet data communications that the significant improvements of the present invention have evolved.

SUMMARY OF THE INVENTION

The present invention, accordingly, advantageously provides apparatus and an associated method, by which more optimally to communicate data packets in a packet communication system, such as a UMTS (Universal Mobile Telecommunications System) wireless data network.

Operation of an embodiment of the present invention better optimizes the size of a transmission window within which a sending station sends a packet of data. By better selecting the size of the transmission window, the throughput rates of data communication of the packet data is improved.

Operation of a further, or alternate, embodiment of the present invention provides a manner by which to adjust a retransmission timer responsive to changes in the characteristics of a radio-link upon which data packets are communicated. The timer is adjusted in a manner to reduce the occurrence of spurious retransmissions as a result of changing radio-link conditions.

In one aspect of the present invention, apparatus is provided for a mobile station, herein referred to as a mobile host, by which to select an optimal transmission window within which a data packet is to be transmitted thereto by a network host. Determination is made at the mobile host of the

optimal size of the transmission window responsive to determination of throughput rates, or other link status indication related to the radio-link. Responsive to the measured, or otherwise determined, indication, selection is made of the optimal transmission window size. A value respective of the optimal transmission window size is then sent to the network host. The optimal transmission window size is used by the network host as a maximum size of the transmission window within which the network host thereafter transmits a data packet.

In another aspect of the present invention, apparatus is provided for a mobile host to select a time-out value for a retransmission timer of the mobile host. RTO estimation is performed in a standard manner. Measurement, or other determination, of a radio-link quality indication is made. Responsive to a value of the radio-link quality being beyond a threshold value. In one implementation, the radio quality indication is representative of changes in communication quality levels. If the changes are significant, the time-out of the retransmission timer is increased. Time-out values which are too low cause spurious time-outs. This results in decreased throughput as congestion control procedures are started, but performance of such procedures is in vain. In a situation in which real congestion of the core network exists, the mobile host does not use an excessively large window.

In one implementation, improved TCP flow control is provided for a mobile host operable in an IP network. Improved throughput rates are made possible

by determinations made at the mobile host of indications related to the TCP communications upon a radio part forming a portion of the communication path formed between the mobile host and a network host (or any sending TCP station). Link layer status information at an RLC layer which specifically provides information about the radio-link is used to optimize communications. Throughput and link status data is used to set an optimal TCP offered window size. And, such data is also used to improve the ability of the retransmission timers of the mobile host to react to changes in the link status or in available bandwidth. In one embodiment, all necessary determinations and selections required for operation of the embodiment of the present invention are effectuated at the mobile host.

In a further implementation, apparatus is provided for a UMTS mobile terminal to optimize better TCP protocols to account for flow and congestion characteristics of wireless communication links. In contrast to conventional TCP flow and congestion control measures which are specifically designed for fixed networks, operation of an embodiment of the present invention takes into account radio-link characteristics to optimize TCP transmission parameters. Data throughput and radio-link status indications, e.g., from a UMTS protocol stack, are used to set an optimal TCP window size. And, the data throughput and radio-link status indications are also used to improve the ability of TCP retransmission timers of the mobile terminal to react to changes in the link status or in available

bandwidth. Implementation of the various embodiments of the present invention can be effectuated entirely at the mobile terminal, thus requiring no changes to fixed network elements, such as IP routers or network terminals.

In these and other aspects, therefore, apparatus, and an associated method, is provided for a first host operable in a communication system in which packet data is communicated between the first host and a second host upon a communication path in which the communication path includes a radio-link. The apparatus, and associated method, selects an optimal window size within which to transmit a data packet. A radio-link status determiner is coupled to receive indications of the radio-link forming a portion of the communication path between the first host and the second host. The radio-link status determiner determines an indication of a characteristic of the radio-link and generates a radio-link status indication indicative of the indication of the characteristic determined thereat. An optimal window size selector is coupled to receive the radio-link status indication generated by the radio-link status determiner. The optimal window size selector selects an optimal window size within which to transmit the data packet.

In these and other aspects, apparatus, and an associated method, are also provided for a communication system in which packet data is communicated between a first host and a second host upon a communication path. The communication path includes a radio-link, and the first host has a

retransmission timer at least for selecting one to retransmit the data packet. Selection is made of a time-out value of a retransmission timer. A radio-link status determiner is coupled to receive indications of the radio-link forming a portion of the communication path between the first host and the second host. The radio-link status determiner determines an indication of radio-link quality of the radio-link and generates a radio-link quality indication indicative of the indication of the radio-link quality determined thereat. A retransmission timer time-out value selector is coupled to receive a value representative of the radio-link quality indication generated by the radio-link status determiner. The retransmission timer time-out value selector selects a time-out value of the retransmission timer. Selection is made responsive to the value representative of the radio-link quality indication.

A more complete appreciation of the present invention and the scope thereof can be obtained from the accompanying drawings, which are briefly summarized below, the following description of the presently-preferred embodiments of the invention, and the appended claims:

BRIEF DESCRIPTION OF THE DRAWINGS

Figure 1 illustrates a functional block diagram of a packet communication system including a mobile host operable pursuant to an embodiment of the present invention.

Figure 2A and 2B illustrate logical layer diagrams illustrating the logical layers of the control and user planes, respectively, of the packet communication system shown in Figure 1.

Figure 3 illustrates a method flow diagram listing the method of operation of an embodiment of the present invention.

Figure 4 illustrates a state diagram representative of an embodiment of the present invention.

Figure 5 illustrates a state diagram representative of another embodiment of the present invention.

DETAILED DESCRIPTION OF THE PREFERRED EMBODIMENT

Referring first to Figure 1, a packet communication system, shown generally at 10, provides for the communication of packet data between a sending station and a receiving station. For purposes of illustration and to describe operation of an embodiment of the present invention, a network host 12 and a mobile host 14 form stations between which packet data is communicated. While the network host 12, in the exemplary illustration of the Figure, is a wireline device, implementation of the network host as a mobile device could alternately be represented. And, while the exemplary implementation shall be described with respect to a TCP (Transmission Control Protocol) network, in which TCP/IP-formatted date packets are communicated between the network and mobile hosts 12 and 14. Other systems can analogously be represented.

Control of when a data packet is transmitted is effectuated during operation of an embodiment of the present invention by taking into account characteristics of a radio-link which forms a portion of a communication path extending between the network and mobile hosts.

The communication system 10 includes network infrastructure, here shown to be formed of an IP-based wireline UMTS (Universal Mobile GPR5 Telecommunications System) core network, including a 3G-GGSN (3G-GPR5 Support Node) 16. The 3G-GGSN 16 is coupled to the network host 12 by way of a packet data network (PDN) 18. The 3G-GGSN is further coupled to a 3G-SGSN (3G Serving GPR5 Support Node) 22. And, the 3G-SGSN is coupled to a radio base station 24. A communication path, here of a wireline nature, extends between the network host and the base station 24 by way of the packet data network, the 3G-GGSN 16, and the 3G-SGSN 22.

A radio-link 26 having both "forward" or "down" link 28 and a "reverse" or "up" link 32 couples the mobile host 14 with the base station 24. The radio-link 26 also forms a portion of the communication path extending between the network and mobile hosts 12 and 14. Other logical structure of the network infrastructure of the system, such as location registers and other nodes are not shown for purposes of simplicity.

The illustrated portions of the communication system, however, illustrate the communication path formed between the network and mobile hosts to include both a wireline core network and a radio

part. Operation of an embodiment of the present invention takes into account the character of the radio-link in controlling flow characteristics of data packets between the network and mobile hosts and also the properties of the fixed line part. In the event of congestion within the fixed-line IP network, normal TCP retransmission is effectuated.

The mobile host 14 includes both a receiver portion 42 and a transmitter portion 44. The receiver portion 42 is operable to act upon data packets received at the mobile host. The transmitter portion 44 is operable to transmit acknowledgments to acknowledge receipt at the mobile host of data packets communicated to the mobile host and also to transmit data packets generated by the mobile host.

The mobile host further includes a controller 48 operable to control operation of the receiver and transmitter portions of the mobile host. The controller includes functional elements of an embodiment of the present invention to select an optimal transmission window size within which the network host communicates data packets to the mobile host.

A retransmission timer 52 functionally represents a timer used to time the transmission of a communication packet during communication of packet data in the communication system 10. The controller further includes a radio-link characteristic detector 54, here coupled to the receiver portion to receive indications of communication characteristics of the radio-link. The radio-link characteristic detector is operable to determine characteristics of the

radio-link therefrom. The controller further includes a flow monitor 56 for monitoring data throughput flows, either on the downlink or uplink paths. The controller also includes an OWND (Optimal Window) calculator 62 coupled to receive indications of determinations made by the radio-link characteristic detector 54. The OWND calculator is operable to calculate an optimal transmission window size of a transmission window within which data packets are to be communicated by the network work to the mobile host. And, the controller further includes a retransmission timer time-out value selector 64, also coupled to the radio-link characteristic detector 54. The retransmission time-out value selector is further coupled to the retransmission timer 52 and is operable to select the time-out value pursuant to which the retransmission timer is operable.

During operation of an embodiment of the present invention, packet data generated by the network host 12 is communicated by way of the communication path to the mobile host 14. Messages communicated to the mobile host include an offered window value, shown at 66, forming a portion of a message communicated to the mobile host. In the exemplary implementation, the contents of the offered window field 72 sent from the mobile host to the network host are affected. The offered window field is a field of a standard TCP header. The flow monitor 56 is operable to provide an indication of data flow rate to the OWND calculator 62. In one implementation, the OWND calculator calculates an optimal window size

responsive to the data throughput rates detected by the flow monitor 56. In a further implementation, the OWND calculator is further operable responsive to detections made by the radio-link characteristic detector 54 in the determination of the optimal transmission window size. Values representative of the calculations performed by the calculator 62 are provided to the transmitter portion 44 which formats the values into an offered window field 72 of a message communicated by the mobile host to the network host. When received by the network host, the value contained in the offered window field 72 forms a maximum transmission window size within which the network host subsequently transmits data packets to the mobile host.

During operation of a further embodiment of the present invention, responsive to detections made by the radio-link characteristic detector 54, the retransmission time-out value selector 64 is selectively operable to alter operation of the retransmission timer 52. In the event that a significant change in the radio-link characteristics is detected, the time-out value of the retransmission timer is increased. When changes in excess of a selected amount are detected, the retransmission timers are prolonged thereby to avoid spurious retransmission due to deterioration of radio-link conditions.

Turning next to Figures 2A and 2B, the communication system 10, shown previously in Figure 1, is again shown, here in logical layer form. The layers of the mobile host 14 are illustrated at the

left-most portion of the Figure (as shown) and, the network portion of the communication system is illustrated at the right-most (as shown) portion of the Figure. Figure 2A illustrates the control plane protocols and Figure 2B illustrates the user plane protocols of the communication system. The logical layers are exemplary of the logical construction proposed for the UMTS system. Other systems can analogously be represented.

In the control plane protocol representation shown in Figure 2A, the mobile host (MS) 14 is shown to include CC and SM layers 76 and 78 which reside upon an MM layer 80. The MM layer resides upon an RRC layer 82 which, in turn, resides upon an RLC-C layer 84. The layer 84 resides upon the MAC layer 86 which resides upon the WCDMA L1 layer 88.

A radio-link indicated by Uu is formed between the mobile host 14 and the radio base station 24. The base station is shown to include a BS-RRC layer 90 and BSAP layer 92. The BS-RRC layer 90 resides upon the MAC layer 94 which, in turn, resides upon the WCDMA L1 layer 96. And, the BSAP layer 92 is shown to reside upon the transport layer 98. The network also includes an RNC 102, here shown to include an RRC layer 104 and an RANAP layer 106. The RRC layer 104 resides upon the RLC-C layer 106, which in turn, resides upon the BSAP and MAC layers 108 and 110 respectively. The layers 108 and 110, in turn, reside upon transport layers 112. And, the RANAP layer 106 resides upon transport layers 114.

The network further is shown to include a CN116, here shown to include CC and SM layers 118 and 120,

corresponding to the layers 76 and 78 of the mobile host. The layers 118 and 120 reside upon the MM layer 122 which, in turn, resides upon the RANAP layer 124. And, the RANAP layer 124 resides upon transport layers 126.

In the user plane representation of Figure 2B, the mobile host 14 is shown to include an L3CE layer 128 residing upon an RLC layer 130 which, in turn, resides upon the MAC layer 132. The MAC layer resides upon a WCDMA/TD/CDMA layer 134. A link, represented by Uu, is formed between the mobile host 14 and an RAN (radio access node) 136. The RAN 136 is shown to include L3CE and GTP layers 138 and 140. The layer 138 resides upon an RLC layer 142 which, in turn, resides upon an MAC layer 144. The layer 144 resides upon the WCDMA/TD/CDMA layer 146. And, the GTP layer 140 resides upon the UDP/TCP layer 148. Layer 148 resides upon an IP layer 150 which, in turn, resides upon a second-level layer L2, 152, and an L1 layer 154.

The network here further shows the logical layers of the 3G-SGSN 22. The node 22 is here shown to include GTP layers 156 and 158. The layers 156 reside upon a UDP/TCP layer 160 which, in turn, resides upon an IP layer 162. The IP layer resides upon L2 layer 164 and, in turn, upon L1 layer 166. Analogously, the GTP layer 158 resides upon a UDP/TCP layer 168, which, in turn, resides upon an IP layer 170. The layer 170 resides upon L2 layer 172 and, in turn, upon L1 layer 174.

Figure 2B further shows the 3G-SGSN gateway node 16. Here, the node 16 is shown to include a GTP

layer 176 which resides upon a UDP/TCP layer 178. The layer 178 resides upon an IP layer 180 which, in turn, resides upon L2 layer 182, and in turn, upon L1 layer 184. It should be noted that, in operation of an embodiment of the present invention, the end-to-end TCP layer is modified rather than the TCP/UDP layer of the core UMTS layer.

Link layer status information is utilized, as noted above, better to optimize TCP. In a first embodiment of the present invention, throughput and link status data available from the UMTS protocol stack, i.e., the layering of data protocols as illustrated in Figures 2A-B is used to set an optimal TCP offered window size.

The mobile host 14 forms a TCP host. During operation of this embodiment, the mobile host is enabled to limit the packet transmission rate of the network host 12 responsive to determinations of the throughput at the radio-link which forms a portion of the communication path between the mobile host 14 and the network host 12. Determination of throughput is conducted on the basis of information obtained from the RLC layers and/or L3CE layers, as appropriate. Control is effectuated over data transmission by the network host 12 to the mobile host 14.

More particularly, the optimal size of a TCP transmission window is selected at the mobile host according to radio-link status information. The selected optimal size, the OWND, is added to the offered window field, also known as the advertised window field, in TCP acknowledgments which are returned to the network host. The network host

utilizes the OWND to define a maximum size of a transmission window within which to transmit a subsequent data packet. Unnecessary congestion and packet losses are avoided as the OWND is chosen according to the wireless link, typically the slowest link in the communication path formed between the network host and mobile host.

The OWND is defined at the mobile host according to information at a UMTS data flow monitor, or QoS monitor. While not separately shown, the monitor is also part of the data protocol stack and is able to give information about the condition of the UMTS network to applications running in the system. Selection is, in one implementation, further responsive to an estimation of the radio-link status. The status is here estimated by measuring round trip-times, by, e.g., measuring the time period required to send an L3CE packet between the mobile host and the radio access network. When radio-link quality deteriorates, or available bandwidth decreases, such time period increases. Changes in this measurement, referred to as the round-trip-time, RTT_{L3CE} are used, for instance to estimate the amount of time required to generate retransmissions over the radio-link. RTT refers to the time interval between sending a packet and receiving an acknowledgment to the sending of the packet. In the event that the L3CE layer does not utilize acknowledgments, the RLC layer RTT is instead utilized

In other words, the TCP layer transmission rates of the network host is limited responsive to the radio-link status. The rate is limited responsive to

calculations of an optimal window size, OWND, at the mobile host by utilizing data throughput information, information related to the time consumed by the RLC layer retransmissions that a L3CE-RTT monitor estimates, and RTTs that the TCP layer estimates. The calculated OWND is inserted into the offered window field of TCP acknowledgments sent from the mobile host back to the network host. In the exemplary implementation, calculations are performed at the mobile host and actual limiting of the data transfer rate is performed at the network host. In other implementations, operation in other manners can be effectuated.

The TCP layer at the mobile host collects user data throughput information provided by a UMTS flow monitor. The receiving TCP layer further utilizes a normal, standardized mechanism to estimate the RTT. Normal operating conditions, e.g., include operating conditions in which throughput in the radio-link is well matched to that required by the fixed TCP/IP network. Retransmission according to the standard TCP retransmission protocol thus maintains protection against packet losses due to congestion in the fixed part of the network.

When a significant change in data throughput is detected, an algorithm is executed to estimate a new value of the OWND. Alternatively, the OWND is regularly calculated and added to an offered window field.

An equation utilized by which to estimate the OWND is as follows:

$$\text{ownd} = \text{Throughput}*\text{RTT}$$

wherein:

OWND is the optimal window size of a transmission window, as calculated at the mobile host;

throughput is the measured data flow rate; and

RTT is the round-trip-time, as defined below.

In the exemplary implementation, the flow monitor, or QoS monitor, monitors throughput in each of the forward and reverse link directions at the L3CE logical layer. And, estimation of the data flow rate towards the mobile host 14 is used as the value of the throughput in the just-listed equation.

RTT estimation, used above, is calculated according to the following equation:

$$RTT = RTT_{TCP} - (RTT_{L3CE} - RTT_{OPT}).$$

wherein:

RTT_{TCP} is an end-to-end round-trip-time calculated utilizing standard TCP mechanisms;

RTT_{L3CE} is a round-trip time at the L3CE logical layer.

RTT_{OPT} is an optimal round-trip-time over the radio-link, i.e., at the L3CE layer.

Following is the manner, in the exemplary implementation, by which the RTT_{L3CE} is measured. When a mobile host is sending an IP packet, the packet is offered to the L3CE layer 128. At such layer, the packet is forwarded to lower layers upon which the L3CE layer resides. At the lower layers, the packet is divided into smaller segments which are transmitted over a radio-link. If corruption occurs, the segments are retransmitted. Once the entire L3CE packet is successfully transmitted, the L3CE layer at

the mobile host is informed by an acknowledgment message transmitted from the receiving host, e.g., a network element of the radio access network. The mobile host is then able to calculate the RTT_{L3CE} by measuring the time interval between the transmission time of the L3CE segment and the arrival time of acknowledgments thereto. The mobile host, could, for example, maintain a table pertaining to recently transmitted L3CE segments. When a segment is passed to a lower layer, the flow to which the packet belongs is identified, e.g., either by a flow ID, or, e.g., identification is made by an IP destination and source address. The TCP destination and source ports and the protocol field, a sequence number and transmission timer is marked in the table. When acknowledgment of the segment is passed from the lower layers to the L3CE layer, the arrival time of the acknowledgment is marked in the table. The segment corresponding to the arrived acknowledgment can then be recognized from the flow ID and the sequence number. The time difference between such markings on the table is the RTT_{L3CE} value. In an implementation in which the L3CE layer does not make acknowledgments, the RLC layer effectuates the acknowledgments. In any event RTT over RAN is measurable at least at one of the layers.

The RTT_{L3CE} value calculated as such, cannot be used to estimate the window size, OWND, as only the time spent in retransmissions over the radio-link should be subtracted from the overall TCP round-trip-time RTT_{TCP}. As a result, RTT_{OPT}, which describes the L3CE layer round-trip time in optimal circumstances,

i.e., exclusive of bit errors and retransmissions, must first be subtracted from the value of RTT_{L3CB}. RTT_{OPT} can be determined by utilizing information that a WCDMA-RLC/MAC layer provides. The RLC/MAC layer, for instance, measures RLC layer round-trip time, which is then used in RLC layer for control. The same information, i.e., RLC layer RTT, can alternately be utilized to calculate the RTT_{OPT} value needed. In another implementation, the value of RTT_{OPT} is determined experimentally at different bandwidths and such experimentally determined values are stored in the mobile host for use in round-trip-time calculations.

The optimal transmission window size, OWND, is then calculated utilizing radio-link status data. The new value of the OWND is then added to an offered window field in a subsequent TCP acknowledgment, or data segment sent from the mobile host to the network host. A network host running in a normal standard TCP protocol handles OWND as it handles normal offered window indications. The network host utilizes OWND as the maximum value for its transmission window. Therefore, there is no need to modify the TCP/IP stack at the fixed host.

Flow control is further effectuated by controlling the manner, and when, TCP retransmission timers are adjusted. The retransmission timer management is in the standard TCP specification and governed according to the following equations:

$$E = RTT_M - RTT_A$$

$$RTT_A \leftarrow RTT_A + gE$$

$$D \leftarrow D + h\ (abs(E) - D$$

$$RTO = RTT_A + 4D$$

wherein:

RTT_M is a most recent RTT measurement;

RTT_A is a smoothed estimator or RTT average;

D is a smoothed mean deviation;

RTO is a value of a retransmission timer; and

h and g are constants of values, e.g., of 0.25 and 0.125, respectively.

In this embodiment, radio-link status information is utilized to identify significant changes in throughput or link quality, and therefrom to adjust the retransmission timers to avoid spurious retransmissions. Furthermore, additional time is provided for standard TCP retransmission timer adjustment, allowing the timer management to settle down to a newly adjusted value. In operation, a value of RTT_{L3CE} is measured for each data packet or segment. When such significant increments in the measured values of round-trip-times (i.e. RTT_{L3CE}) are detected, the information about the change in the values is passed on to a TCP layer. The RTO, above-defined, of each TCP segment currently on-transmission is prolonged thereby to avoid spurious retransmissions. Spurious TCP retransmissions are decreased in situations in which the mobile host sends data and either the available bandwidth over the radio-link or the radio-link quality deteriorates significantly. Link layer status information is utilized in the timer adjustment. When deterioration is detected, the TCP retransmission timer values are prolonged. This is done both to the TCP segments that have already been transmitted and the timer is

currently running as well as to at least a subsequent segment to be next transmitted. In this manner, the network has more time to effectuate retransmissions at the lower layers, (e.g., the RLC layers) prior to timing out of the TCP timers, resulting in retransmissions of TCP segments. It should be noted that prolonging an RTO does not affect RTT estimations. The timers may be extended for one or more packets or segments still remaining to be transmitted.

An RTT_{L3CE} estimation table, described above, is also utilized to detect changes in the radio-link. When the quality of the link or radio resource is available for the TCP flow decreases, values of the RTT_{L3CE} estimations increase. As the table includes history information of RTT_{L3CE} measurements such changes are detectable. The difference, between, for instance, the two most-recent RTT_{L3CE} measurements is the time interval noted in the retransmission timer adjustment.

When the mobile host is to send a TCP segment, the RTT_{L3CE} table is accessed to determine whether there has been a significant change between the latest measurements of the TCP data flow. If there has not been a change, the TCP resets the retransmission timer in a conventional manner. If, however, the status of the radio-link has changed, the difference between the latest RTT_{L3CE} measurements of that specific data flow, T_D is calculated and added to a TCP retransmission timer.

A significant change between successive measurements can be determined, e.g., by measuring a

deviation D in the previously determined RTT_{TCP} values which may be calculated as shown in the previous equations. If T_D is bigger than, e.g., 4D, a significant change is defined, and the TCP timers are prolonged. Examination of the effect of different choices on the number of spurious TCP retransmissions under given conditions might result in a selection of another value.

T_D is also added to the timers of the TCP segments that have already been transmitted. To do so, the TCP implementation must be such that if the retransmission timer times-out, retransmission of TCP segments is not done unconditionally. Instead, a check is made to see whether there is a need to prolong the timer. If there is, a new timer with a value T_D is started and, if this timer also times-out, a normal standardized retransmission procedure is effectuated. In the event that there is no need to prolong the original timer, a normal retransmission procedure is performed subsequent to expiration of the timer.

When a significant change in T_D is detected, the timers of the segment to be transmitted subsequently and also N following segments are prolonged. To determine the proper value of N the constants h and g in the preceding equations can be used. Such constants determine how strong the impact of the most recent measurements of RTT and deviation D have when setting the value of RTO. Consequently, N is chosen so that after N segments, the normal RTO estimate has enough time to settle down to the new value and timers need not be prolonged by T_D anymore. A proper

value of N can, for instance, be experimentally determined, such as by setting N to 1/h.

Figure 3 illustrates a method, shown generally at 212, of an embodiment of the present invention. The method is operable to select an optimal window size within which to transmit a data packet. First, and as indicated by the block 214, an indication of throughput quality on the radio-link is determined. Then, and as indicated by the block 216, a radio-link status indication indicative of the throughput quality is generated. And, as indicated by the block 218, an operable window size within which to transmit a data packet is selected.

Figure 4 illustrates a state diagram, shown generally at 224, representing the manner by which retransmission timers are adjusted during operation of an embodiment of the present invention. A first state 226 represents L3CE data transmission and reception. A path is taken from the state 226 to a time table 228. Packet transmission time and reception time of a corresponding acknowledgment arc written on the table together with corresponding flow ID (or IP addresses and TCP port numbers and protocol ID). And, paths are taken from the time table 228 to an RTT_{L3Ce} calculation state 232 and through an RTT_{L3CE} change detection state 234. To calculate the RTT_{L3CE} values, the table is accessed to read transmission and reception times. The result of the calculation is written to the table. To detect the change in the RTT_{L3CE}, the table is accessed and the latest RTT_{L3CE} values are collected. The change to T_D, is passed forward. Once changed detection is determined at the

state 234, a path is taken to a delay selection state 236. TD and D are used to determine whether there is a need to prolong the timers. If readjustment is needed, the number of subsequent TCP packets to be affected, i.e., N, is determined using g and/or h, and T_D and N are passed forward. Paths also extend to the standard TCP RTO estimation state 238. The standard TCP and RTO estimation functionality passes forward essential information. Such information includes standard deviation D and constants g and h. And, paths extend from the states 236 and 238 to a TCP retransmission timer control logic state 242. The TCP retransmission timer control logic controls timers. If readjustment is required the control entity readjusts timers currently running by passing a value of new prolonging timer forward, and also N following segments by adding TD to the RTO_{TCP} to form RTO. If there is no need for readjustment, the standard TCP RTO estimate is utilized The path extends from a state 242 to a TCP/IP data packet transmission and reception and timers state 244. TCP/IP data transmission and reception entity exchanges data with the L3CE transmission and reception entity. Retransmission timers controlled at the state 244 are used to time the transmission. Timer implementation is such that if there is a need to prolong a running timer, it can be done by starting a new one right after the running timer expires. Also, TCP layer RTT measurements are passed to the RTO estimation state 238 to run standard RTO estimation. RTT_{TCP} measurements are returned to the state 238 on a path formed therebetween.

Figure 5 illustrates a state diagram, shown generally at 252, representing the manner by which to estimate a window (OWND) during operation of an embodiment of the present invention. A first state 254 represents L3CE (or RLC) layer data transmission and reception. Packet transmission time and reception time of a corresponding acknowledgment are written on a time table 256 together with corresponding flow ID (or IP addresses and TCP port numbers and protocol ID).

At an RTT_{L3CE} calculation state 258, RTT_{L3CE} values are calculated. To calculate such values, the table 256 is accessed to read transmission and reception times. The result of the calculation is written to the table. And, at an RTT calculation state 260, calculations performed at the state 258 and stored at the table 256 are used to calculate values of RTT.

The state diagram also includes a UMTS flow monitor state 262. The flow monitor state 262 receives flow information from the state 254 and uses the information to generate required information. L3CE layer throughput information is passed forward to the OWND calculation state 264. Also, if the standard UMTS flow monitor state is able to give an indication of the radio-link status, it can be used to provide RTT_{OPT} information to the state 258. The OWND is calculated at the state 264 using throughput and RTT information. The resulting OWND size is passed to the TCP.IP transmission/reception state 266.

A standard TCP RTO estimation state 268 is provided to estimate RTO in normal manner. RTO is

passed forward to the TCP/IP transmission/reception entity that also includes timer control and timers. TCP layer RTT estimation, i.e., RTT_{TCP}, is also passed to the RTT calculation state 258.

At the state 266, data is exchanged with the state 254. OWND information obtained from the state 264 is used as a maximum value of the TCP transmission window and the OWND is also added to TCP packets or acknowledgments sent towards the other host. TCP layer RTT measurements are passed to the RTO estimation entity to run the standard RTO estimation.

Thereby, operation of an embodiment of the present invention permits improvement in packet communications upon a communication path including a radio-link. By taking into account the conditions of the radio-link to control data flow, congestion which might otherwise occur as a result of packet losses on the radio-link is reduced. An embodiment of the present invention is further implementable to effectuate mobile-to-mobile communication by way of a fixed, e.g., UMTS, network. The above-described apparatus and method, in such an implementation, is utilized at both radio-links extending to both of the mobile hosts.

The previous descriptions are of preferred examples for implementing the invention, and the scope of the invention should not necessarily be limited by this description. The scope of the present invention is defined by the following claims:

We claim:

1. In a communication system in which packet data is communicated between a first host and a second host upon a communication path, the communication path including a radio-link, an improvement of apparatus for selecting a window size within which to transmit a data packet, said apparatus comprising:

a radio-link status determiner coupled to receive an indication of the radio-link forming a portion of the communication path between the first host and the second host, said radio-link status determiner for determining an indication of a characteristic of the radio-link and for generating a radio-link status indication indicative of the indication of the characteristic determined thereat; and

a window size selector coupled to receive the radio-link status indication generated by said radio-link status determiner, said window size selector for selecting a window size within which to transmit the data packet.

2. The apparatus of Claim 1 wherein the indication of the radio-link to which said radio-link status determiner is coupled to receive comprise indications of a packet data throughput rate upon the radio-link.

3. The apparatus of claim 2 wherein the radio-link includes a forward link and a reverse link and wherein the indication of the packet data throughput rate is of at least a selected one of the forward link and the reverse link.

4. The apparatus of claim 3 wherein the indications of the radio-link to which said radio-link status determiner is coupled to receive comprises indications of a RTT (round-trip-time) representative of a time period within which two-way transmission with the first host are determined to be effectuated.

5. The apparatus of claim 4 wherein the communication system is defined by logical layers, the logical layers including an RLC (radio-link control) layer, and wherein the time period of which the RTT is representative corresponds to a round-trip time of communication of a data packet and an acknowledgment thereto less retransmission times at the RLC layer.

6. The apparatus of claim 4 wherein the optimal window size selected by said optimal window size selector is proportional to the indication of the data throughput rate and to the RTT.

7. The apparatus of claim 1 wherein the first host comprises a mobile host and the second host comprises a network host and wherein the mobile host is further operable to send a message to the network host, wherein the message includes an offered window field, and wherein a value representative of the optimal window size selected by said optimal window size selector is inserted into the offered window field.

8. The apparatus of claim 7 wherein the network host is operable to receive the message, including the value representative of optimal window size, sent by the mobile host, and to transmit the data packet, the data packet transmitted by the network host within a transmission window of a maximum size corresponding to the optimal window size.

9. The apparatus of claim 1 wherein the first host further comprises a retransmission timer at least for selecting when to retransmit a data packet and wherein said radio-link status determiner is further for determining an indication of a characteristic of the radio-link forming a portion of the communication path and for generating a radio-link quality status indication indicative of the indication of the characteristic determined thereat, a value representative of which is applied to the retransmission timer and responsive to which a time-out value of the retransmission timer is selected.

10. The apparatus of claim 9 wherein the radio-link status indication generated by said radio-link status determiner is representative of changes in the radio-link status between a first selected time and at least a second selected time.

11. The apparatus of claim 10 wherein, if the radio-link status indication is beyond a selected value, the time-out value of the retransmission timer is caused to be increased.

12. The apparatus of claim 11 wherein the communication system is defined by logical layers, the logical layers including an L3CE layer, and wherein the radio-link status indication comprises a RTT (round-trip-time) representative of a time period within which two-way transmissions with the first host at the L3CE layer are determined to be effectuated.

13. The apparatus of claim 12 wherein the time-out value of the retransmission timer is caused to be increased if changes in values of the RTT exceed a multiple of a mean deviation.

14. The apparatus of claim 12 wherein the time-out value of the retransmission timer is caused to be increased if more than a selected number of data packets are communicated upon the communication path without acknowledgment thereto.

15. The apparatus of claim 12 wherein the time-out value of the retransmission timer is caused to be increased during transmission of a selected number of data packets subsequently to be transmitted.

16. The apparatus of claim 1 wherein the first host and the second host are operable to communicate data packets pursuant to a TCP (transmission control protocol) protocol and wherein said radio-link status determiner determines the indication of the throughput quality of TCP-formatted data packets.

17. In a communication system in which packet data is communicated between a first host and a second host upon a communication path, the communication path including a radio-link, and the mobile host having a retransmission timer at least for selecting when to retransmit a data packet, an improvement of apparatus for the first host for selecting a window size within which to transmit the data packet, said apparatus comprising:

a radio-link status determiner coupled to receive indications of the radio-link forming a portion of the communication path between the first host and the second host, said radio-link status determiner for determining an indication of a characteristic of the radio-link and for generating a radio-link quality indication indicative of the indication of the characteristic determined thereat; and

a retransmission timer time-out value selector coupled to receive a value representative of the radio-link status indication generated by said radio-link status determiner, said retransmission timer time-out value selector for selecting a time-out value of the retransmission timer responsive to the value representative of the radio-link status indicators.

18. In a method for communicating packet data between a first host and a second host, the first host and the second host connected together by way of a communication path which includes a radio-link, an improvement of a method for the first host for selecting a window size within which to transmit a data packet, said method comprising:

determining an indication of throughput on the radio-link;

generating a radio-link status indication indicative of the throughput determined during said operation of determining; and

selecting a window size within which to transmit the data packet responsive to a value of the radio-link status indication.

19. The method of claim 18 wherein the first host further comprises a retransmission timer at least for selecting when to retransmit a data packet and wherein said method further comprises the operations of:

determining an indication of radio-link status of the radio-link forming a portion of the communication path;

generating a radio-link status indication indicative of the indication of the radio-link status determined during said operation of determining; and

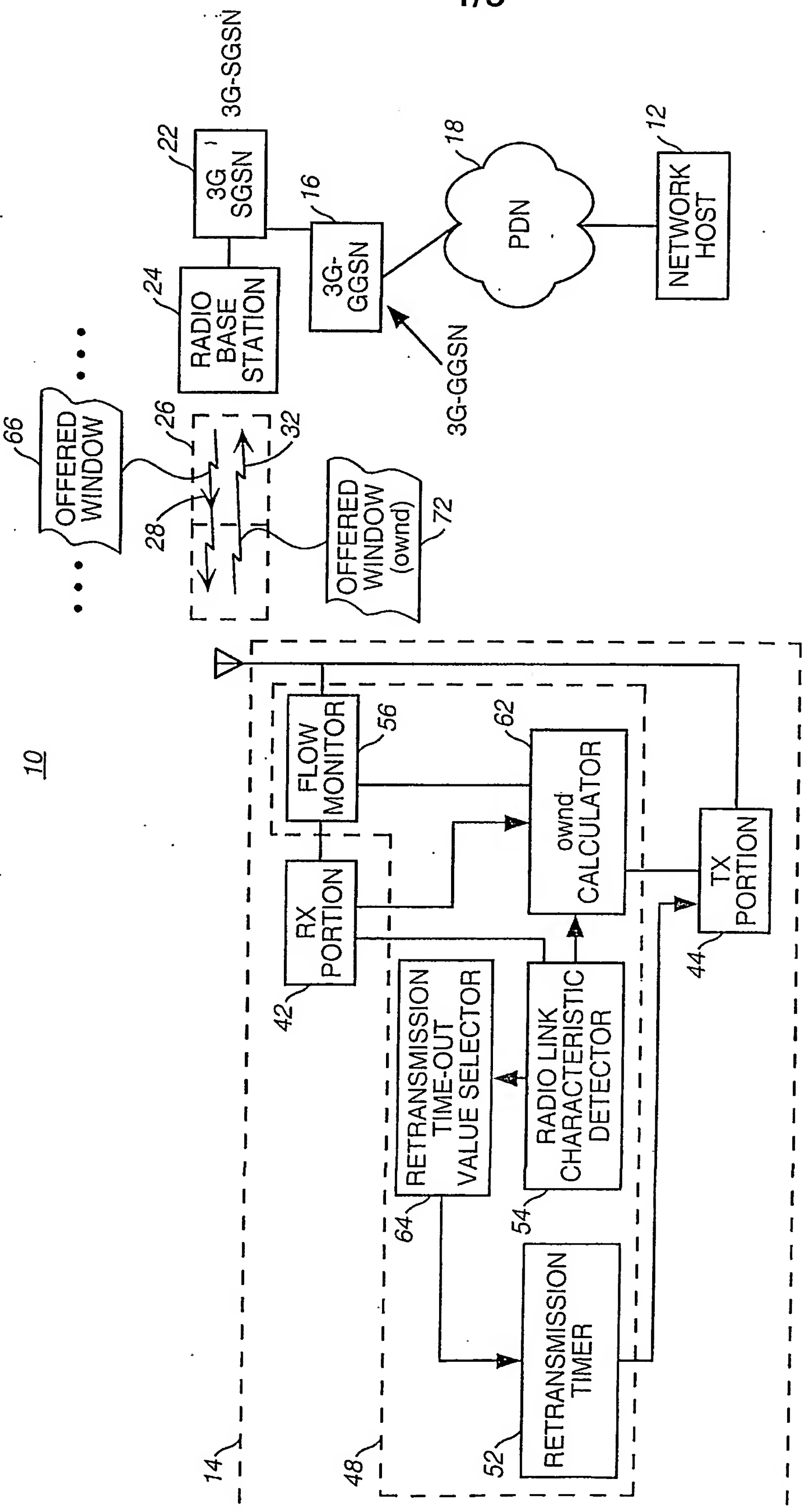
adjusting a time-out value of the retransmission timer if the radio-link status indication is beyond a selected threshold.

20. The method of claim 19 wherein the radio-link status indication determined during said operation of determining the indication of radio-link status is representative of changes in the radio-link status.

21. The method of claim 19 wherein the indication of the radio-link status determined during said operations of determining the indication of the

radio-link status comprise an indication of a packet data throughput rate.

22. The method of claim 21 wherein the indication of the data throughput rate comprises a RTT (round-trip-time) representative of a time period within which two-way transmissions are effectuated.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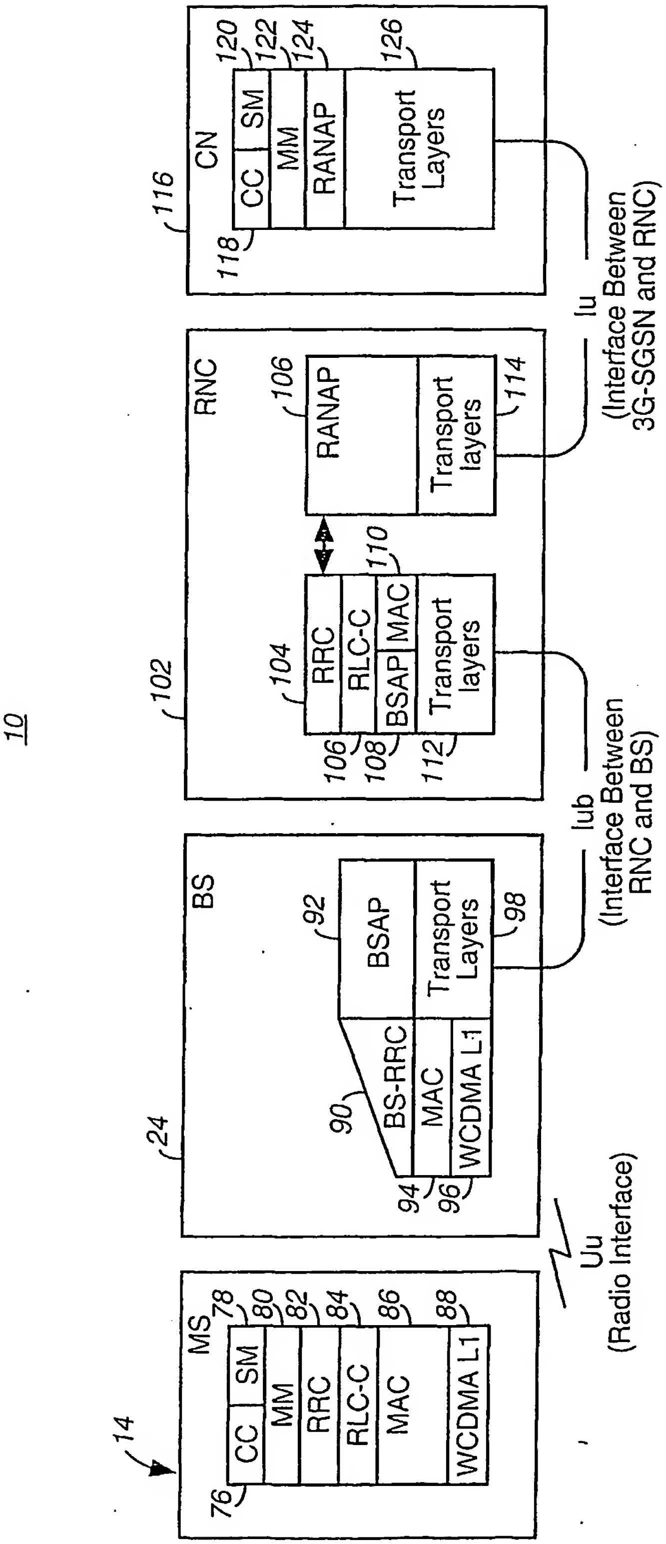
22
3G SGSN
24
RADIO BASE STATION
16
3G-GGSN
3G-GGSN
18
PDN
12
NETWORK HOST
66
OFFERED WINDOW
26
28
32
OFFERED WINDOW (ownd)
72
14
48
FLOW MONITOR
56
62
ownd CALCULATOR
RX PORTION
42
TX PORTION
44
RETRANSMISSION TIME-OUT VALUE SELECTOR
64
RADIO LINK CHARACTERISTIC DETECTOR
54
RETRANSMISSION TIMER
52

FIG. 1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118
120
122
124
126
CC
SM
MM
RANAP
Transport Layers
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RNC
102
104
106
108
110
112
114
RRC
RLC-C
BSAP
MAC
Transport layers
RANAP
Iub (Interface Between RNC and BS)
BS
24
90
92
94
96
98
BS-RRC
BSAP
MAC
WCDMA L1
Transport Layers
Uu (Radio Interface)
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MM
RRC
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MAC
WCDMA L1

FIG. 2A

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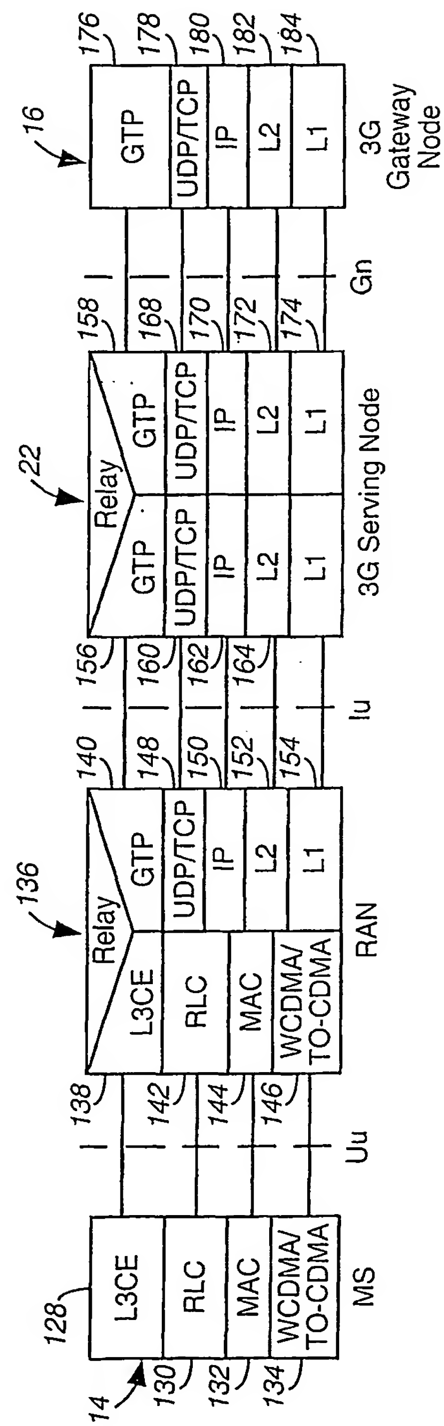
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RLC
MAC
WCDMA/ TO-CDMA
RAN
Relay
L3CE
RLC
MAC
WCDMA/ TO-CDMA
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UDP/TCP
IP
L2
L1
3G Serving Node
Relay
GTP
UDP/TCP
IP
L2
L1
GTP
UDP/TCP
IP
L2
L1
3G Gateway Node
GTP
UDP/TCP
IP
L2
L1
Uu
Iu
Gn
14
128
130
132
134
136
138
140
142
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146
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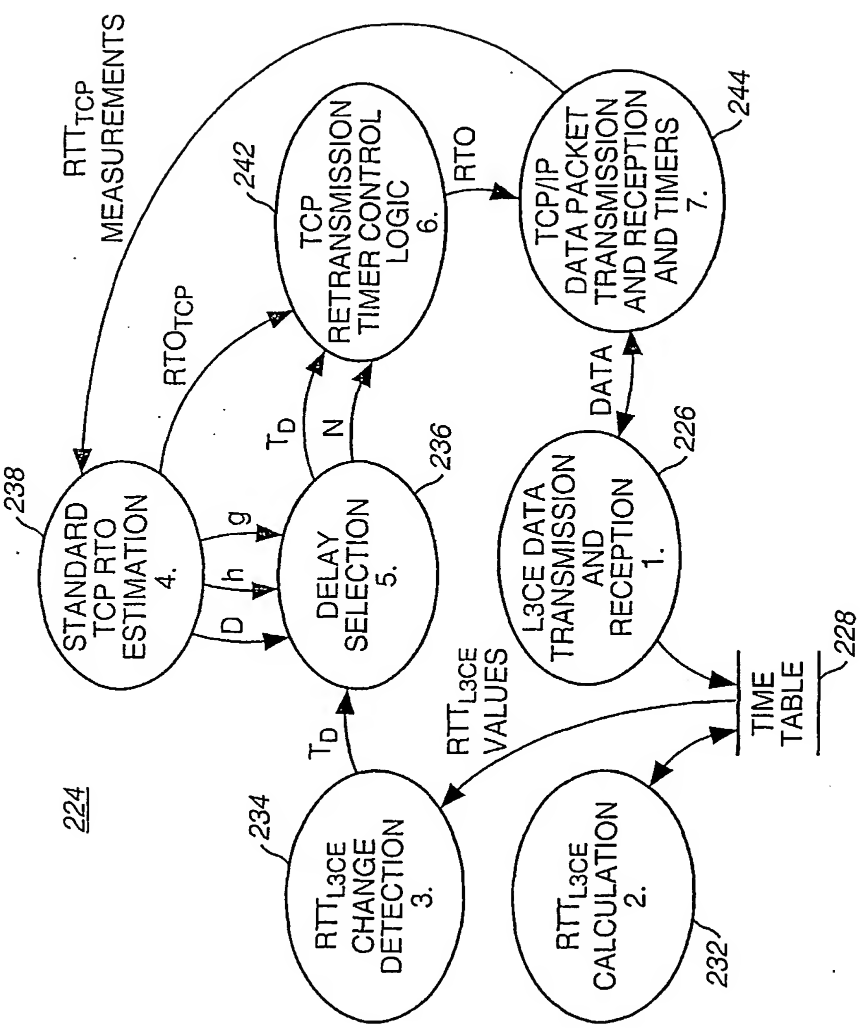
FIG. 2B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RTT$_{TCP}$ MEASUREMENTS
STANDARD TCP RTO ESTIMATION 4.
238
RTO$_{TCP}$
TCP RETRANSMISSION TIMER CONTROL LOGIC 6.
242
RTO
TCP/IP DATA PACKET TRANSMISSION AND RECEPTION AND TIMERS 7.
244
D
h
g
DELAY SELECTION 5.
236
T$_D$
N
DATA
L3CE DATA TRANSMISSION AND RECEPTION 1.
226
RTT$_{L3CE}$ CHANGE DETECTION 3.
234
RTT$_{L3CE}$ VALUES
TIME TABLE
228
RTT$_{L3CE}$ CALCULATION 2.
232

FIG. 3

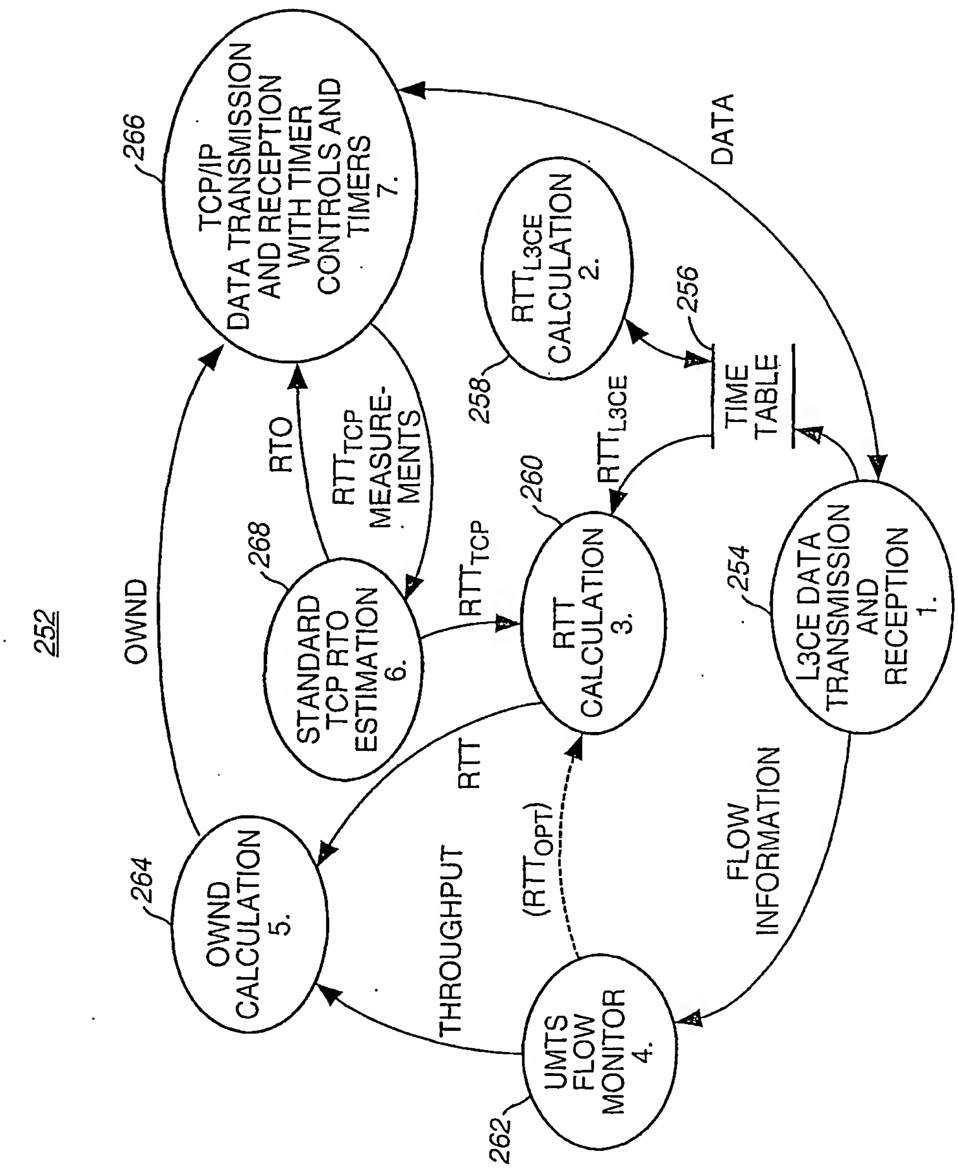
252
TCP/IP DATA TRANSMISSION AND RECEPTION WITH TIMER CONTROLS AND TIMERS 7.
266
DATA
RTT$_{L3CE}$ CALCULATION 2.
258
256
TIME TABLE
RTT$_{L3CE}$
OWND
RTO
RTT$_{TCP}$ MEASURE-MENTS
268
STANDARD TCP RTO ESTIMATION 6.
RTT$_{TCP}$
260
RTT CALCULATION 3.
254
L3CE DATA TRANSMISSION AND RECEPTION 1.
264
OWND CALCULATION 5.
RTT
THROUGHPUT
(RTT$_{OPT}$)
FLOW INFORMATION
UMTS FLOW MONITOR 4.
262

FIG. 4